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COMMUNICATION SYSTEMS—SPECIAL

MONOGRAPH B-594

SOME ACOUSTICAL PROBLEMS OF SOUND PICTURE ENGINEERING

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A Review of Several Acoustical Problems of Sound Picture Engineering Selected to Illustrate the Recent Trend of Advancement in Acoustics as Applied to Sound Pick-up

Presented at

ANNUAL CONVENTION OF
THE INSTITUTE OF RADIO ENGINEERS
CHICAGO, ILL. JUNE, 1931

Published in

PROCEEDINGS OF THE INSTITUTE OF RADIO ENGINEERS
VOL. 19, PP. 1606-1614 SEPTEMBER, 1931

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Printed in the United States of America

SOME ACOUSTICAL PROBLEMS OF SOUND PICTURE ENGINEERING*

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Summary—*The purpose of this paper is to point out that many advances in acoustical engineering have been necessary in order to understand and control adequately the conditions under which modern sound pictures are recorded and reproduced. To illustrate this point, some of the acoustical problems encountered at Bell Telephone Laboratories are discussed. The sudden and successive changes in sound intensity level to be expected in a room during the growth and decay of sound from an intermittent source are pointed out. The necessity of using the more general reverberation time formula, which was developed over a year ago, when dealing with comparatively "dead" rooms, is indicated. One type of acoustical distortion which is due to interference is discussed together with the measures necessary to minimize it in sound pick-up work. These phases of acoustical engineering have been selected for discussion from many which confront the engineer in this field.*

THE development of the technique of sound picture recording and reproduction has presented many acoustical problems. Most of these have been concerned with improvements in the quality of recording and the illusion of reality of the reproduced sound. It was necessary to consider what acoustical conditions were required on motion picture sets for the proper recording of the sound. These involve the design and construction of the sets and the acoustical treatment which is necessary on the sound stage in which the sets are built and also the technique of proper sound pick-up by the microphone. On the other hand many theatres have been treated with acoustical materials in such a way as to reduce the reverberation time and eliminate objectionable echoes. Many of these problems are similar to those which are encountered in broadcasting and for this reason it may be desirable to review some of them.

For many years there has been an established science of acoustics which has furnished a basis for the correction of unsatisfactory acoustical conditions in many auditoriums and the proper design of others. Such a scientific background has been extremely useful in that it has served to encourage effective corrective measures and to discourage many methods of acoustical treatment which are not based on principles which are sound from an engineering standpoint. As an example of this latter type of treatment, one recalls the rather well-established idea that a multitude of wires strung overhead produce satisfactory listening conditions.

* Decimal classification: 621. 385. 96. Original manuscript received by the Institute, April 22, 1931. Presented before Sixth Annual Convention of the Institute, June 5, 1931.

In an attempt to apply this established science of acoustics to these modern problems, however, one of the first things which became obvious was that the older methods of describing acoustical phenomena were inadequate in that they did not give a sufficiently detailed and accurate picture of the situation. In order to illustrate this let us choose for discussion a few of the many acoustical problems which confront the sound picture engineer.

First, let us consider the growth and decay of sound energy in an enclosure. Fig. 1 illustrates one method of plotting the average sound intensity throughout a room against the time for a sound source which is turned on at the time $t=0$ and cut off at the time $t=0.2$. As will be seen on inspecting the curves the average sound intensity increases very rapidly at first and then much less rapidly as the average intensity

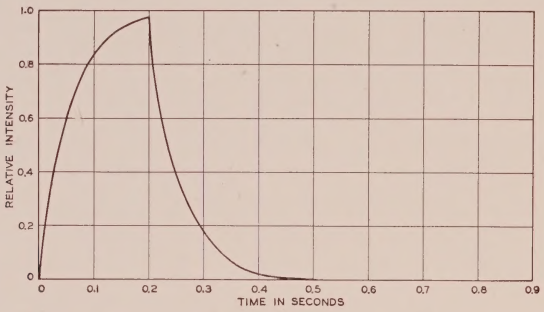

Fig. 1—Relative intensity of sound (averaged throughout the room) vs. time in seconds measured from the time the source begins to operate.

reaches the maximum value which is attained after the sound has been on long enough to establish a steady state condition. If after the steady state condition has been reached the sound source is cut off suddenly, the average sound intensity in the room will decrease in accordance with the curve shown to the right of the abscissa, $t=0.2$.

If the data shown in Fig. 1 are replotted using a logarithmic scale for the sound intensity the result is the curve shown in Fig. 2. The unit often used in this case is the decibel which is well known in acoustical circles. Since the sense of hearing exhibits approximately a logarithmic response to the stimulus of sound, this curve represents very nearly the rise and fall of the intensity level as the ear hears it, and therefore it suits our purpose better to plot the data in this way. As may be seen, the intensity level reaches a point which is within one decibel of maximum in a surprisingly short time and the decay of the average intensity level is uniform with time, that is, the plot of the decay is a straight line. The slope of this straight line indicates the rate of decay of the

sound intensity level. In a room, of given size and shape, this rate depends on the amount of sound absorption present. Since undue persistance of sound is a frequent cause of poor hearing conditions considerable importance is attached to the study of the rate of decay of sound intensity level in auditoriums. It has become conventional to describe this property of a room by stating the time it takes the sound intensity to decay to one millionth of its value after the sound source is cut off (after a steady state condition has been established), or using the scale of sound intensity level shown in the last figure, it is the time it takes the sound intensity level to decrease 60 decibels. This interval of time is called the reverberation time.

In sound picture engineering, and especially in recording, we are concerned with the variation in the intensity level at any one point in a

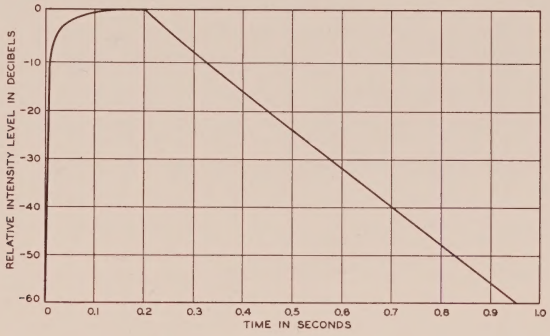

Fig. 2—Relative intensity level (averaged throughout the room) vs. time. The data shown in Fig. 1 are here plotted to a different scale of ordinates.

room. Immediately, we realize that this established method of approach is inadequate. For instance, at a point ten feet from a sound source the increase in the sound intensity would not take place the instant the source is sounded as might be inferred from the above figures but would be delayed until the sound waves have traveled the intervening distance. When this direct wave arrives there is an abrupt increase in the sound intensity followed by a further sharp increase when the reflection from the floor arrives. Fig. 3 illustrates what happens.[1] Here again, the abscissas are the intervals of time after the sound source has been started and the ordinates indicate the sound intensity level at any time. As is seen, the intensity level increases very rapidly at the moment the direct wave from the source arrives and it increases

[1] This method of analysis has been applied by R. L. Hanson of the Bell Telephone Laboratories to calculating the changes in sound intensity level at discrete points in sound picture studios and sets.

further in small increments as the floor reflection and the multitude of reflections from the walls arrive. In a very short time, however, the intensity level is very near to its maximum value. If the sound source is cut off at the time $t=0.2$, it is again a short interval of time before the end of the direct wave passes by, causing the first change in the sound intensity. There is another change when the train of waves reflected from the floor passes by and further discrete changes as the wave trains from other surfaces pass by. But soon there are so many of these and each makes such a small change in itself that the decay becomes essentially uniform with time. This part of the decay is illustrated on the curve by a straight line. Considering the sound pick-up from this viewpoint we see clearly the rapid and intermittent changes in sound intensity which are taking place.[2]

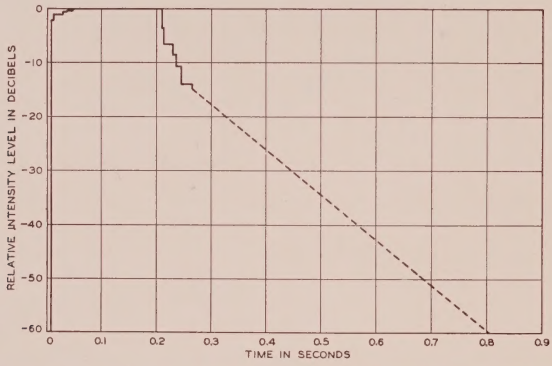

Fig. 3—Relative intensity level at a point ten feet from the source vs. time.

This last picture of the building up and decay of sound intensity level has been constructed for a simple room. In practical cases many irregularities are present. The plane surfaces of the walls may be interrupted by doors and windows. Usually the room contains furniture and there are people present. These objects will more or less scatter and diffuse the sound waves and the more they are diffused the less abrupt will be the changes of intensity level so prominent in Fig. 3.

This discussion of the rise and fall of the sound intensity in enclosures will serve to point out that the science of acoustics must be ad-

[2] These changes have been illustrated in Fig. 3 as successive decreases in the sound intensity level. This plot has been made without regard to the relative phases of the various wave trains. If these are taken into consideration, the magnitude, but not the time, of any particular change may be different from that illustrated. In fact, sometimes the passing of a wave train is indicated by an increase in level. This is the case when destructive interference is eliminated by the cessation of one of the interfering waves. On the whole, however, the in tensity level decreases rapidly, essentially as illustrated.

vanced and extended if it is to serve the demands of modern engineering adequately.

For many years engineers concerned with applied acoustics have been active in measuring, calculating, and controlling the reverberation time in auditoriums and other enclosures. With the advent of radio broadcasting and sound pictures, activity in this field has been multiplied. Therefore, a correct reverberation time formula which may be used in designing rooms to have predetermined acoustical characteristics is much to be desired.

Until recently the formula used for this purpose is as follows:

$$T = \frac{0.05V}{a} = \frac{0.05V}{S\alpha_a}$$

in which T is the reverberation time in seconds,
V is the volume of the room in cubic feet,
a is the number of absorption units in square feet,
S is the surface of the room in square feet, and
α_a is the average coefficient of absorption of the surface.

The upper curve in Fig. 4 shows the reverberation time at various fre-

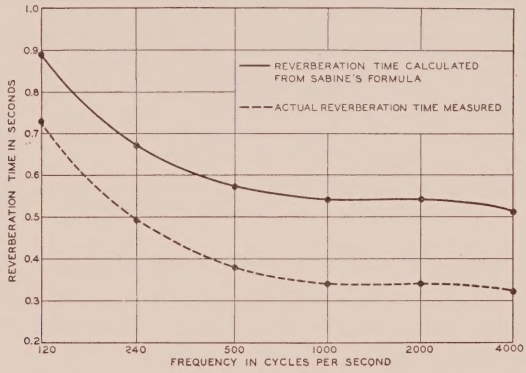

Fig. 4—Reverberation time vs. frequency for a comparatively "dead" studio.

quencies, calculated by means of this formula for the sound stage of Bell Telephone Laboratories. After the room was completed, however, equipment was available to actually measure the reverberation time at various frequencies with the results shown by the other curve on Fig. 4. It will be seen the discrepancy between the calculated and measured reverberation time throughout the whole frequency range is quite large, amounting to 50 per cent for some frequencies. Such a disparity between predicted and actual results can be quite disturbing. In this case

the solution of the problem proved extremely valuable. An extremely careful analysis of the derivation of this reverberation time formula indicated that it is applicable to "live" rooms only and that it is not suitable for rooms which are comparatively "dead" as sound stages usually are.

On the basis of these careful considerations, Eyring[3] has developed a more general formula which is applicable to all rooms whether "live" or "dead." This formula is

$$T = \frac{0.05V}{-S \log (1 - \alpha_a)}.$$

Comparing this to the older formula it will be noticed that the change has been to put $-\log_e(1-\alpha_a)$ in place of α_a. For "live" rooms, where the average absorption coefficient is small, the two formulas yield results

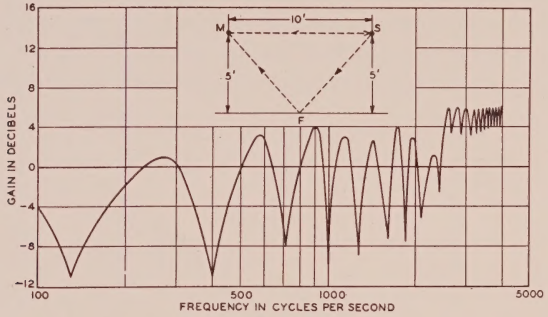

Fig. 5—Transmission frequency characteristic between sound source, S, and microphone, M, located as shown.

which are practically identical, but in the case of dead rooms where α_a is comparatively large the newer formula gives much lower values of the reverberation time than the older one and these lower values agree with those actually measured experimentally.[4]

Another acoustical problem of sound picture engineering is concerned with the amount of distortion which may be introduced into the recording system between the speaker and the microphone. Unless precautions are taken, this cause of impaired quality of the sound record may prove serious. One of the main causes of this type of distortion is the interference at the microphone of two or more sound waves com-

[3] C. F. Eyring, *Jour. Acous. Soc. of Am.*, **1**, 217–241; January, 1930. Fig. 4 has been taken from this paper.

[4] It is believed by many that there is a limited range of reverberation time which is optimum for recording in any one studio. See, for instance, Maxfield and Harrison, *Bell Syst. Tech. Jour.*, **5**, 493–523; July, 1926.

ing from the same source by different paths. A quantitative analysis of this type of distortion has been made by Hanson.[5] Fig. 5 shows the transmission frequency characteristic of sound from a source S to a microphone M situated as illustrated in the sketch. In this case the direct wave travels ten feet to the microphone while the wave reflected from the floor travels about fourteen feet.

The explanation of this phenomena is simple. Since the sound waves from the source reach the microphone by two paths, one of which is longer than the other, there will be certain frequencies for which the two waves arrive in phase and there will be other frequencies for which the two waves will arrive exactly 180 degrees out of phase. On this basis we can plot a theoretical frequency characteristic of the transmission which appears in Fig. 6. The peaks of the curve appear at those frequencies for which the two waves are in phase and the valleys appear

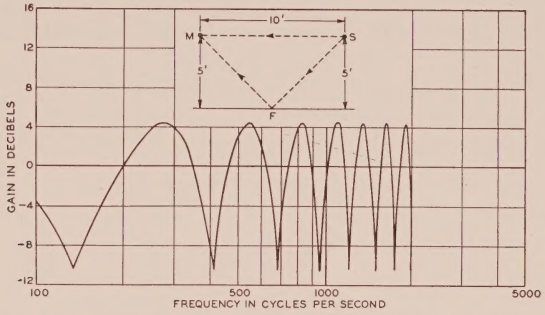

Fig. 6—Theoretical transmission frequency characteristic between sound source, S, and microphone, M, located as shown, assuming the reflecting surface to have a reflection coefficient of 0.95.

at these frequencies for which the two waves appear out of phase and partially cancel one another. It is noticed that the frequencies at which the peaks and valleys of the experimental curve occur are very nearly the same as those at which the peaks and valleys of the theoretical curve appear.

After covering the floor with very efficient absorbing material and adding a second microphone placed about fourteen feet from the source, the transmission characteristic between the source and the combined outputs of the two microphones was found to be as illustrated in Fig. 7. In this case there are two paths which the sound can take to reach the electrical recording system and the lengths of these two paths are exactly the same as those in the previous case considered, namely, ten

[5] R. L. Hanson, *Jour. S.M.P.E.*, **15**, 460–470; October, 1930. Figs. 5 to 8 have been taken from this paper.

and fourteen feet. Note that here also the peaks and the valleys come at the same frequencies at which the peaks and valleys occur on the theoretical curve. The explanation of the curve then is the same as before except that the interference takes place between the electrical outputs of the two microphones.

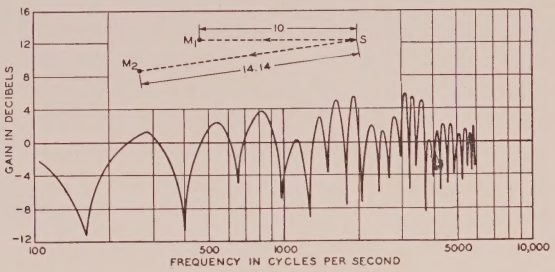

Fig. 7—Transmission frequency characteristic between a source and the combined outputs of two microphones located as shown.

Experimental sound records made under conditions where distortion due to interference existed show a very characteristic unnatural quality which is described by some as "hollow." Conditions of recording

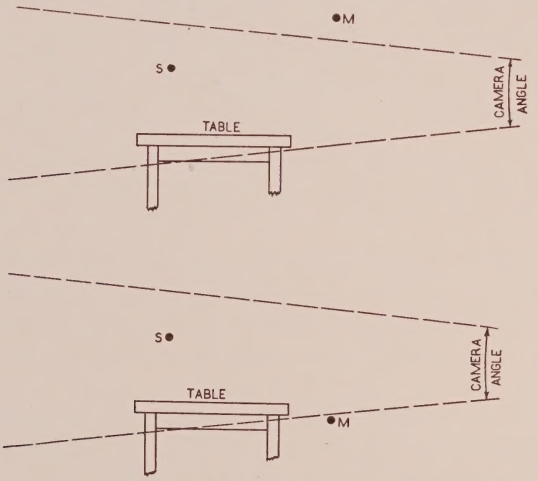

Fig. 8—The upper illustration shows an arrangement of speaker *S*, and microphone, *M*, which would result in a sound record of poor quality due to the interference at *M* of the direct sound and that reflected from the table. The lower illustration indicates a location of the microphone such that it receives the direct sound only.

which often lead to this distortion are found in cases where a speaker is talking near a table, a wall, or a mirror in such a way that the sound

striking the near-by surface is reflected into the microphone. One such case is illustrated in Fig. 8. The arrangement of speaker *S* and microphone *M* in the upper part of the figure illustrates a condition which would result in a sound record of poor quality due to the interference at the microphone of the direct sound and that reflected from the table. However, the position of the microphone shown in the lower part of this figure is such that the sound reflected from the table top cannot reach it. This arrangement would result in a sound record which is not affected by this form of distortion.

Acoustic distortion of this type can be avoided by the elimination of reflecting surfaces in the immediate proximity of the speaker or by a proper selection of the microphone position so that it will avoid sound reflected from these surfaces.

We have discussed the growth and decay of sound intensity in an enclosure indicating the necessity of considering the intermittent changes which occur more or less suddenly. We have seen also the necessity of using the more general reverberation time formula in modern acoustical work. The possibility of severe acoustical distortion which may occur between the source of sound and the microphone has been called to your attention. These phases of the acoustical problems encountered in sound picture engineering at the Bell Telephone Laboratories have been discussed with a view to pointing out that many advances in acoustical engineering have been necessary in order to understand and control adequately the conditions under which modern sound pictures are recorded and reproduced.

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